REMARKS

This paper is in response to the Office Action of February 4, 2009, the term to respond extends to June 4, 2009, with one month extension.

Request for Interview:

A request for Interview is hereby made. If the Examiner feels that the claims are not in condition for allowance, the Inventor has indicated his availability to discuss the merits of his technical claims. From the inventor's review, the claim are believed to be patentable, and the prior art of record fails to teach or suggest his now claimed embodiments. Thus, before another action is issued by the Office, the undersigned respectfully requests the opportunity to discuss the case in a joint interview the inventor of this present application.

Amended Claims:

The newly amended independent claims 7, 25, and 31 direct the claimed invention to embodiments described in the as-filed application. Specifically, the independent claims were amended to focus on one specific embodiment, which does not limit the scope of other disclosed embodiments, which may be claimed in a continuation application, if desired. For each of understanding, the Applicant points out the following elements, which have been integrated into the independent claims, and which differentiate the claimed invention over the cited art. Namely, the processing using blind source separation/second order statistics, enables the system to work in a noisy environment with reverberation.

As claimed, the periodic calibration, by way of background monitoring, provides users with a freedom to move around, changing the position of the controller relative to the target speaker/user. The prior art fails to teach or suggest the newly claimed combination. The features added by amendment include:

- **Blind Source Separation/Second Order Statistics**: As now claimed, the independent claims recite that the calibration uses blind source separation and second order statistics to separate the target signal and the noise. See, as filed specification, paragraphs [0040, 0041, 0043, 0046, 0049].
- **Separation/Frequency Basis**: Since the time-delay between background and foreground from various sensors is different, their second-order statistics in

frequency spectrum domain are independent of each other, therefore, the signals may be separated on a *frequency component basis*. See Paragraph [0051]. As now claimed, the calibration implements blind source separation and second order statistics to separate the target audio signal from the noise based on a frequency basis.

- **Both first and second Filters Updated**: As claimed, the periodically setting of the calibration is done for both a value of the first filter and a value of the second filter based upon the monitored acoustic set-up, which occurs in the background.

- **Calibration is Periodic**: As now claimed, the calibration remains fixed between the periodic setting. As such, the background monitoring can occur during operation, but the calibration is only periodically updated. Thus, the target signal component is able to freely move around in 3-dimensional space with six degrees of freedom relative to the microphone array of the game controller. See Paragraph [0028]

Support was listed by paragraph, in the as-filed specification. Thus, the newly amended claims do not introduce new matter. In view of the clarifying amendments, the Applicant respectfully request reconsideration.

Rejections Under 35 USC § 103

Claims 7-9, 11-13, and 25-34 were rejected under 35 USC § 103(a) over Yang et al. (7,206,418) in view of Best (4,305,131). This rejection is respectfully traversed.

Yang et al. is newly cited art, as the Applicants arguments overcame the previous rejections. In light of the amendments made herein, the new rejections are respectfully traversed.

Yang et al. generally teaches methods for suppressing noise from a signal, where speech and noise reside. Yang et al. uses microphones that can be placed on devices, such as cellular phones, PDAs and other electronic devices that are designed to receive speech. In the primary embodiment of Yang et al., certain microphones are used for voice input, and are designed at a location where the user's mouth might reside. Col. 3, line 64. In addition, Yang et al. teaches a method that actively detects "activity". Thus, the first beam forming usnit is allowed to adapt during the active time periods, and the second beam forming unit is allowed to adapt during non-active time periods. Col. 3, lines 1-5; Col. 5, lines 52-61.

Importantly, the processing taught by Yang et al. requires that a beam forming controller 218 to provide the "necessary" control to dictate when each beam former is to adapt, **separately**. Thus, adaptation takes place at different times, for *both* the first and second filters. See Figs. 3A and 3B of Yang et al. This is key for Yang et al., as it attempts to *only* adapt to noise when speech is silent, and then adapt to the speech when it is determined to be occurring, based on the "voice activity detector" 440. The adaptation of Yang et al. is therefore fundamentally different than the claimed *calibration of* the *both* first and second filters, using blind source separation that implements *second order* statistics.

Furthermore, the adaptation described by Yang et al., defines the use of an adaptation algorithm, such as LMS (least mean square), to determine the theoretical upper bound on how much noise can be cancelled using linear adaptation filter, such as blocks 420 and 550. Col. 10, lines 44-56. Yang et al. therefore teaches how to use "*linear filters and linear transfer functions*" for adaptation of the filters. Col. 9, lines 57-61. And further, the adaptation of the first and second filters occurs as ***different times***, thus causing uneven adaptation. This is not equivalent to the claimed periodic calibration of both the first and second filters, based on background monitoring. Although Yang et al. says "non-linear" functions may be used, no working example is provided--and thus no teaching. Col. 9, lines 60-61, and Col. 12, lines 5-6. Even in the example that was so called non-linear (col. 11, line 47 to col. 12, line 4), the result is a "*linear combination*."

Still further, by way of the claimed embodiments, the user is not directing its voice toward the microphones *per se*, but instead, the controller is designed to feely move in space in various interactive directions (during interactivity), which may or may not be in front of the user's mouth. Yang et al. requires some directionality or at least expects a user to speak toward the device. Col. 3, line 64; Col. 4, line 33; Col. 12 line 54 (noting that certain microphones are defined to receive the desired speech from the user).; Col. 12, line 66 (microphone 110b/Fig. 1B; being close to the mouth);

Best fails to cure the deficiencies of Yang et al. The teachings of Best disclose a gaming control with a microphone. In particular, Best fails to disclose a second filter performing reverse beam-forming to track the target signal component and whose coefficients

are adjusted depending on a changing acoustical set-up to steer the second filter toward the target signal component.

The combination of Yang et al. and Best under §103 fails to disclose each element of the newly amended independent claims 7, 25, and 31. The dependent claims were also amended, and reconsideration is respectfully requested.

If the Examiner has any questions concerning the present amendment, the Examiner is kindly requested to contact the undersigned at (408) 774-6903. If any fees are due in connection with filing this amendment, the Commissioner is also authorized to charge Deposit Account No. 50-0805 (Order No SONYP028).

Respectfully submitted,
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